International Journal of Engineering and Applied Sciences (IJEAS)
ISSN: 2394-3661, Volume-2, Issue-2, February 2015

Survey On Acoustic Echo Cancellation Based On Adaptive Filter

Kusum Sahu, Rahul Sinha

***Abstract*— In digital signal processing an Adaptive filtering constitutes one of the core technologies and finds numerous application areas in science as well as in industry. In wide range of applications Adaptive filtering techniques are used, including echo cancellation, adaptive equalization, and adaptive noise cancellation. An adaptive filter is a system with a linear filter that has a transfer function controlled by variable parameters and a means to adjust those parameters according to an optimization algorithm. Echo is the reflected copy of the voice heard some time later and delayed version of the original.In telecommunication system Acoustic Echo cancellation are used. When interference of the signal occurred by Acoustic Echo, then it discomposes to the user and thus the quality of communication is reduces. Echo cancellers are very successful and today almost no echo at all can be perceived while using telephones. In this paper RLS algorithm was implemented for the echo cancellation.By using this algorithm MSE (mean square error) can be reduced, convergence rate can be improved and thus increasing the communication quality.**

***Index Terms*— Acoustic echo cancellation, Adaptive filtering, MSE (Mean Square Error), RLS algorithm.**

I. INTRODUCTION

An adaptive filter is a computational device that attempts to model the relationship between two signals in real time in an iterative manner. According to an adaptive algorithm adaptive filter self-adjusts the filter coefficients. Below figure shows the diagram of an adaptive filter.

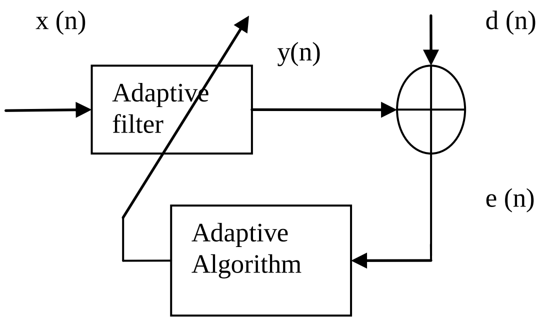

Fig1: Block diagram of Adaptive Filter

Where the input signal to a adaptive filter is represented by x (n)

The output signal is represented by y (n)
An error signal that denotes the difference between d (n) and y (n) is given by e (n)

In signal processing, when signals and transfer functions are not varying with time, then fixed filters are designed. But in many applications both signals and transfer functions are varying with time; in this case fixed filters are not used, so a self adjusting filter is used. This self adjusting filter is called as Adaptive filter. However Mendhe S.K. et al. (2014) proposes the adaptive echo cancellation using normalized least mean square (NLMS) algorithm. The NLMS algorithm is the normalized version of least mean square (LMS) algorithm. Today's technical scenario communication system possesses additive noise, signal interference and echo etc. Due to this reason the error is generated at the time of transmission of data. Hence adaptive filter is a appropriate option to reduce the noise or channel effects. In this paper high speed of adder, subs tractor and Adaptive filter coefficients to design LMS and NLMS algorithm is realized. For reducing the unwanted echo and for increasing the communication quality this paper gave two algorithm LMS and NLMS. The NLMS algorithm establishes a better balance between simplicity and performance than least mean square algorithm [1]. Dhiman J. et. al. (2013) describes the comparison between adaptive filtering algorithms that is least mean square (LMS), Normalized least mean square (NLMS), Recursive least square (RLS). Three performance criteria are used in the study of these algorithms: the minimum mean square error, the algorithm execution time and the required filter order. It concludes that according to SNR improvement the best Adaptive Algorithm is the RLS (recursive least square) algorithm. This algorithm exhibits fast convergence and less steady state error compare to both LMS and NLMS algorithm [2]. Yathiraju R. (2013) focuses on the use of Least Mean Square (LMS), Normalized Least Mean Square (NLMS), Variable Step-Size Least Mean Square (VSLMS), Variable Step-Size Normalized Least Mean Square (VSNLMS) and Recursive Least Square (RLS) algorithms for removing unwanted echo. It concluded that LMS algorithm is most popular algorithm because of its simplicity but in real time application NLMS algorithm is used due to its better performance when compared to LMS algorithm. In this paper the author also concluded that VSNLMS algorithm has poor performance and RLS algorithm has greater algorithm compared to all other algorithm [3]. Arya S. et al.(2011) proposed a multi sub filter approach for echo reduction and comparing their results. This paper is organized in four sections. Section two describes the simulation model of AEC in mat lab using multifactor approach. Further, section three discusses the results. In the end section four concludes the paper. This system is based upon a least-mean-square (LMS) adaptive algorithm and uses multi filter technique. Thus MSE results show that single long adaptive filter shows poor performance as compared to multiple sub filter structure [4]. Homana I. et al (2009) presented a scheme for the acoustic echo canceller. Several adaptive algorithms are presented and simulated to choose the best implementation. Because of its simplicity, the LMS algorithm is the most popular adaptive algorithm. The NLMS algorithm, an equally simple, but more

Kusum Sahu, M.E. Scholar, Electronics & telecommunication, CSIT Durg (C.G.), India
Rahul Sinha, Assistant Professor, Electronics & telecommunication, CSIT Durg (C.G.), India

robust variant of the LMS algorithm, exhibits a better balance between simplicity and performance than the LMS algorithm. Due to its good properties the NLMS has been largely used in real-time applications. The RLS algorithm has the greatest attenuation of all studied algorithms and converges much faster than the LMS algorithm. Due to the large number of multiplications it is rather costly to be implemented [5]. Tundon A. et al. (2006) concluded that NLMS algorithm has simplest architecture and it gives the better performance so it is widely used in industrial areas for echo cancellation. This algorithm will become costly when it is used for multichannel system. To overcome this problem partial update of the filter coefficients are used [6].

II. ACOUSTIC ECHO CANCELLATION

Echo is the reflected copy of the voice heard some time later and delayed version of the original. In order to improve voice quality on a telephone call Echo cancellation is used in telephone to describe the process of eliminating echo from a voice communication system. Echo cancellation is the process which removes unwanted echoes from the signal on a telephone line. It includes first detect the actual transmitted signal that re-appears, with some delay, in the transmitted or received signal. It can be removed by 'subtracting' this echo from the transmitted or received signal, when the Echo is detected.

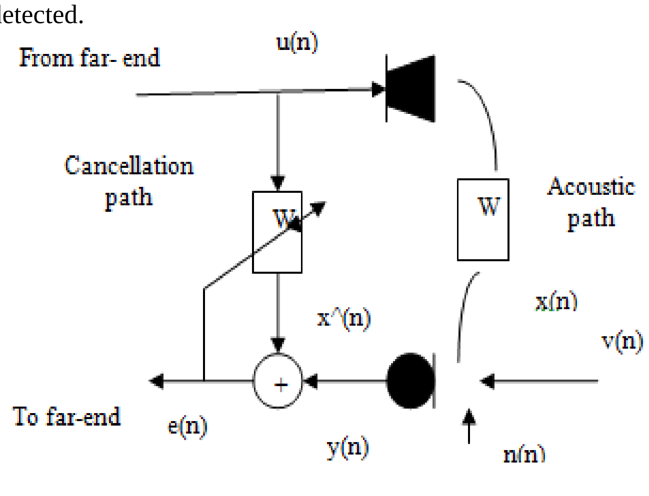

Fig2: Set-up for AEC

In telecommunication systems Acoustic echo cancellation is used. It is used in a hands-free loudspeaker telephone, which operate in a full duplex mode. When interference of the signal occurred by Acoustic Echo, then it discomposes to the user and thus the quality of communication is reduces. When an audio signal is reverberated in the environment then Acoustic echo occurs, then we get the original intended signal plus attenuated or time delayed images of the original signal. In this paper the occurrence of acoustic echo in telecommunication systems is focused. This system consists of both the acoustic input and output devices, which are active concurrently. A hands-free telephony system is an example of this input and output devices. In this scenario the system has both the microphone input and an active loudspeaker operating simultaneously.

Then the system behaves as both a transmitter and receiver in full duplex mode. When a signal is received by the microphone input, then it is output through the loudspeaker into an acoustic environment. Loudspeaker output signal is reflected through the Acoustic environment and returned to the system via the microphone input. The time delayed images of the original signal is contained by these reverberated signal, which are then returned to the original transmitter. By this Acoustic Echo the signal interference occurred and thus the communication quality will be reduces.

III. RLS ALGORITHM

In this RLS algorithm is used for Acoustic Echo Cancellation. An algorithm which repeatedly finds the filter coefficients that reduces a weighted linear least squares cost function relating to the actual input signals. This algorithm differs from other algorithms such as the least mean squares (LMS) that aim to reduce the mean square error. From the derivation of the RLS, the input signals are considered deterministic, while for the LMS and some other similar algorithm they are considered stochastic.

RLS was discovered by Gauss but lay unused or ignored until 1950 when Plackett rediscovered the original work of Gauss from 1821. The RLS can be used to solve any problem that can be solved by adaptive filters. For example, suppose that a d(n) signal is transmitted over an echoey channel , noisy channel that causes it to be received as

$$x(n) = \sum_{k=0}^{q} b_n(k)\, d(n-k) + v(n) \qquad (1)$$

where $v(n)$ represents additive noise. We will attempt to recover the desired signal $d(n)$ by use of a $p+1$-tap FIR filter, :$\mathbf{w}$

$$\hat{d}(n) = \sum_{k=0}^{p} w_n(k)\, x(n-k) = \mathbf{w}_n^T \mathbf{X}_n \qquad (2)$$

Where $\mathbf{X}_n = [x(n)\ x(n-1) \ldots \ldots x(n-p)]^T$ is the vector containing the $p+1$ most recent samples of$x(n)$. Our goal is to predict the various parameters of the filter $\mathbf{w}$, and at each time *n* we refer to the new least squares estimate by $\mathbf{w}_n$. As time evolves, we would like to avoid completely removing the least squares algorithm to find the new estimate for $\mathbf{w}_{n+1}$, in terms of $\mathbf{w}_n$.

IV. METHODOLOGY

In this Section the Acoustic Echo Cancellation by using RLS algorithm in RLS Filter is designed using MATLAB simulink model and the analysis of result will be done. In this input of the RLS filter is given by the sine wave including noise signal in addition with the Echo signal which is the delayed version of the original signal. Our desired signal should be the original signal without noise, which is given to the second input of the RLS filter.

+The difference of desired Signal and Input Signal is the error Signal which is obtained from error Signal output port of RLS block. The desired signal is taken from the output of the RLS filter.

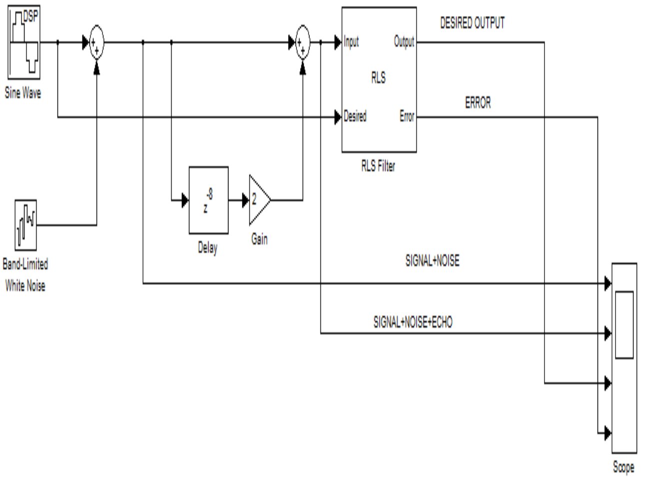

Fig3. Simulink model for Acoustic Echo Cancellation

Input signal

Here the sine wave block generates a real or complex sine wave signal .it has following parameters which have been applied for getting the result.

- Amplitude-2
- Frequency-50 Hz
- Phase offset(rad)-0
- Sample mode-Discrete
- Output complexity-Real
- Computation method-Trigonometric function
- Sample time-1e^{-3}
- Sample per frame-1
- Resetting states when re-enabled-Restart at time zero
- Output data type-Double
-

Band limited White Noise

This noise is added with the input signal and it is applied to the RLS filter with Echo signal. The purpose of our filter is to suppress the Echo signal with the noise. There parameters are

- Noise power- 0.5e^{-4}
- Sample time-1e^{-3}

RLS filter

There are various types of adaptive algorithms are available .Here we are using RLS algorithm for Acoustic Echo cancellation. There parameters are-

Parameter Description:-

- Algorithm-RLS
- Filter length-32
- Forgetting factor(0 to1)-1.0
- Initial value of filter weights-0
- Initial input variance estimate-0.1

V. RESULT

The result for simulink model for Acoustic Echo Cancellation is shown in fig 4. In this, the first waveform represents the input signal with noise. Second waveform represents the Echo signal with input signal and noise. The third waveform represents the desired output signal. The fourth waveform is shows the error signal which defined by the difference of desired signal and input signal. Here we observe a slight variation in desired output signal. The error signal represented is approximately zero value. So the RLS filter is best option for Acoustic Echo cancellation.

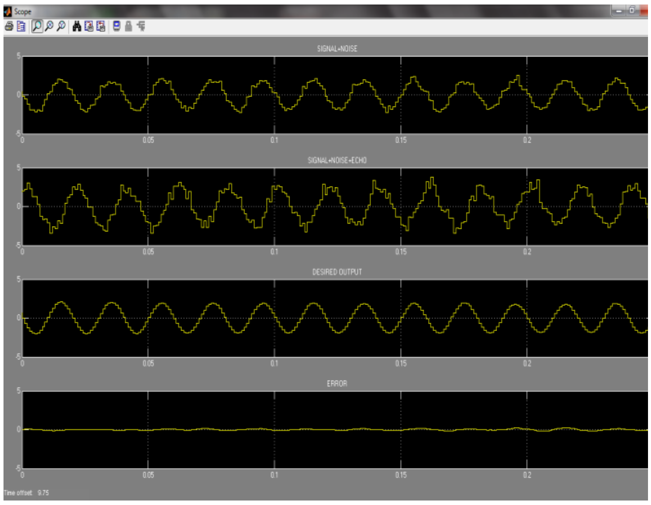

Fig4. Result for Simulink model

VI. CONCLUSION

Adaptive Digital Signal Processing is a special branch of DSP, dealing with adaptive filters and system design. There are number of adaptive algorithms available in literature review and every algorithm has its own properties, but aim of every algorithm is to obtain a minimum mean square error (MSE) at a higher rate of convergence with lesser complexity. Thus algorithms RLS algorithm are having better performance is chosen. The results for the RLS algorithm are given in fig4. From this plots it can be shown that the results obtained for echo cancellation are the best for the RLS algorithm. The estimation error is very small, even smaller than the NLMS algorithm. Though the RLS algorithm gives much better results compared to other algorithms still it is not used, as each iteration requires 4N^2 multiplications. For echo cancellation systems the FIR filter order is usually in the thousands. Thus the number of multiplications required is very large because of which the RLS algorithm is too costly to implement. In practice the LMS based algorithms, although poorer performers are preferred.

VII. FUTURE SCOPE

In the future the Simulink model using XILINX software is designated and simulation result analysis can be done for comparing the result of MATLAB Simulink model and XILINX software model. Then After hardware cost is also calculated.

REFERENCES

[1] Mendhe S.K., Chede S.D. "*Design and Implementation of Adaptive Echo Canceller Based LMS & NLMS Algorithm*" in IJAIEM Volume 3, Issue 6, June 2014.

[2] Dhiman J., Ahmad S.," *Comparison between Adaptive filter Algorithms (LMS, NLMS and RLS)*" in International Journal of Science, Engineering and Technology Research (IJSETR) Volume 2, Issue 5, May 2013.

[3] Yathiraju R. '' *Acoustic Echo Cancellation Using Conventional Adaptive Algorithms*" in International Journal of Inventive Engineering and Sciences (IJIES) Volume-1, Issue-12, November 2013.

[4] Arya S. , Sahu O.," Performance *Evaluation of Adaptive Filters Structures for Acoustic Echo Cancellation*" in International Journal of Engineering (IJE), Volume (5) : Issue (2) : 2011.

[5] Homana I." *Echo cancelling using adaptive algorithms*" in IEEE 2009.

[6]Tundon A." *An efficient, low-complexity, normalized lms algorithm for echo cancellation*", in IEEE 2006.
[7] Dhamija S., Tayal A. *"Adaptive filtering algorithms for channel equalization and echo cancellation"* in IJCTA 2011.
[8] Per Åhgren "*Acoustic Echo Cancellation and Doubletalk Detection Using Estimated Loudspeaker Impulse Responses*", in IEEE 2005
[9] Christof Faller and Jingdong Chen," *Suppressing Acoustic Echo in a Spectral Envelope Space",* in IEEE 2005.
[10] Kumar M. "*Comparative Analysis of various Adaptive Filtering Algorithms for Adaptive System Identification*", in IJCET 2014
[11] Matthias Pawig" *Adaptive Sampling Rate Correction for Acoustic Echo Control in Voice-Over-IP*",in IEEE 2010.
[12] Harsha I. K. Rao "*Fast LMS/Newton Algorithms for Stereophonic Acoustic Echo Cancelation*", in IEEE 2009.
[13] Danilo Comminiello" *Functional Link Adaptive Filters for Nonlinear Acoustic Echo Cancellation*",in IEEE 2013
[14] Sheng Wu, Xiaojun Qiu, and Ming Wu" *Stereo Acoustic Echo Cancellation Employing Frequency-Domain Preprocessing and Adaptive Filter*", in IEEE 2011
[15] Jacob Benesty "*A Hybrid Mono/Stereo Acoustic Echo Canceler*",in IEEE 1998.
[16] Jose M. Gil-Cacho "*A frequency-domain adaptive filter (FDAF) prediction error method (PEM) framework for double-talk-robust acoustic echo cancellation*",in IEEE 2014


Kusum Sahu, received BE degree in Electronics & Telecommunication from the Chhattisgarh Swami Vivekanand Technical University, INDIA in 2012. Presently studying in ME(Communication) from the Chhattisgarh Swami Vivekanand Technical University, in Department of Electronics & Telecommunication, Chhatrapati Shivaji Institute of Technology, DURG (CG).


Rahul Sinha, received the ME (Communication) from the Swami Vivekanand Technical University, INDIA in 2013 and BE degree in Electronics & Telecommunication from the Pt. Ravishankar Shukla University, INDIA in 2005. He has done Advanced PG Diploma in VLSI Design in 2006. He has served as an ASIC engineer in a private organization for more than two years. Presently working as Assistant Professor in Department of Electronics & Telecommunication, Chhatrapati Shivaji Institute of Technology, DURG (CG). His research interest includes signal processing & VLSI